

PTDI

1. A company with a 56 kbps data network link between two locations wants to pass two concurrent VoIP calls between locations. The customer priority is the best possible voice quality to meet call requirements and network capacity. Which CODEC best meets the customer's network requirements?

64

$$\frac{56 \text{ kbps}}{2} = 28 \text{ kbps}$$

IM 64Kps
729 (8K)
726 (10K)
723

- A. G.711
- B. G.723.1
- C. G.726
- D. G.729A

PTDI

2. Given the following parameters: (exhibit button)

- Voice Activity Detection (VAD) is disabled
- CODEC is G.711
- Voice packet payload is 20 ms → 1600
- 100 Mbps Ethernet switched LAN network

$$\frac{10000}{20} = 500$$

18 = 18ms, 18K

Approximately how many IP phones can make simultaneous calls if Call Admission Control (CAC) limits VoIP bandwidth to 10 Mbps?

- A. 64
- B. 100
- C. 128
- D. 156

$$3W = (160 + 40 + 18) \times 8 \times 50 = 87200$$

$$\# = \frac{10,000,000}{87200} = 114.68 \rightarrow 115$$

PTDI

3. A customer is planning to install a VoIP network and has bandwidth constraints on the existing data link. The customer wants to use less bandwidth, but maintain good voice quality. Which CODEC and voice packet payload should you recommend to meet this customer requirement?

- A. G.726 with 10 ms of payload
- B. G.726 with 20 ms of payload
- C. G.729 with 20 ms of payload
- D. G.729 with 30 ms of payload

BW QoS

BW mela QoS

PTDI

4. Which three call volume and voice packetization parameters should you use to determine bandwidth requirements for a VoIP solution? (Choose two)

- A. CODECs
- B. Jitter buffer
- C. Voice sample
- D. Echo cancellation
- E. Voice Activity Detection (VAD)

PTDI

5. When comparing the G.107 E-Model to the Mean Opinion Score (MOS), which characteristic does NOT accurately describe the G.107 E-Model?

- A. Its scale is typically from 50 to 94
- B. It is the numerical average of the results from all listeners

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Pay 21

- C. It models-packets loss distribution and end-user perception
 D. It can be used to express end-to-end voice quality measurement in a customer environment

PTDI 6. Which two CODECs have the least processing or algorithm delay? (Choose two)

- A. G.711
 B. G.723.1
 C. G.726
 → D. G.729

PTDI 7. A company wants to achieve the most efficient use of bandwidth over its 128 kbps ...
 ... The company's priority is the best possible voice quality to meet call requirement (two concurrent calls) and network capacity. Which CODEC best meets the customer's network?

- A. G.711
 B. G.723.1
 C. G.726
 → D. G.729A/B

$$\frac{128 \text{ kbps}}{2} = 64 \text{ kbps}$$

PTDI 8. A company with a 56 kbps data network link between two locations wants to pass two concurrent VoIP calls between locations. The customer priority is the lowest bandwidth consumption, NOT voice quality. Which CODEC best meets the customer network requirements?

- A. G.723.1
 B. G.726
 C. G.729A
 D. G.764

$$\frac{56 \text{ kbps}}{2} = 28 \text{ kbps}$$

PTDI 9. What is the total voice quality using of these combined MOS scores? (exhibit button)

- A. Bad
 B. Poor
 → C. Fair
 D. Good
 E. Excellent

$$\begin{array}{r} 27 \overline{) 8} \\ 30 \overline{) 33} \\ 6 \end{array}$$

PTDI 10. Given the following parameters: (exhibit button)

- Voice Activity Detection (VAD) is disabled
- CODEC is G.729
- Voice packet payload is 20 ms → 20 Kbytes
- 12 simultaneous calls → 50pps

What should be the appropriate bandwidth to use for the fractional Frame Relay T1/E1?

- E. 128 kbps
 F. 256 kbps
 → G. 364 kbps

$$(20 + 40 + 6) \times 8 \times 50 = 26400$$

$$12 \times 26400 = 316800$$

316.8 kbps

B. 512 kbps

$$\frac{384 \text{ Kbps}}{2} = 192 \text{ Kbps}$$

PTDI 11. Given the following company network information:

- Additional T1/E1 (384 kbps) is running Point-to-Point Protocol (PPP) between two sites
- There is no data on the line
- There are 24 IP phones at each site using G.711 CODEC on the LAN
- The VoIP gateway to the WAN (fractional T1/E1) is using G.729 CODEC to reduce bandwidth
- The customer reports that the voice quality degrades when there are five or more calls active
- Due to the network architecture calls from the IP phones through the VoIP gateways are incurring transcoding

$$24 \text{ Kbps} \times 5 = 120 \text{ Kbps}$$

What are the four individual recommendations you should discuss with the customer to improve voice quality?

- A. Add bandwidth, deploy QoS, change CODECs, and limit the number of calls
- B. Add bandwidth, deploy QoS, change T1/E1 formatting to Frame Relay (FR), and limit the number of calls
- C. Increase Maximum Transmission Unit (MTU), deploy QoS, change CODECs, and limit the number of calls
- ✗ D. Increase Maximum Transmission Unit (MTU), deploy QoS, change T1/E1 formatting to Frame Relay (FR), and limit the number of calls

ADNV 12. Before beginning a network assessment of a customer's data network for VoIP, why is it important to understand the customer's perception of "good or acceptable" voice quality?

- A. Understanding the customer's determination of good/acceptable voice quality will affect the CODEC(s) selected for the network
- B. Understanding the customer's determination of good/acceptable voice quality will have minimal impact on the assessment on the network
- C. Understanding the customer's determination of good/acceptable voice quality will affect the understanding of the network traffic requirements
- D. Understanding the customer's determination of good/acceptable voice quality will affect the prioritization of voice over data traffic on the network

ADNV 13. A company requires a 30 day assessment performed on its network. Which two tools are best suited for gathering the necessary data to ensure a successful VoIP deployment without requiring an engineer onsite for the duration of the testing? (Choose two)

- A. Sniffer Pro
- B. HP Openview
- C. NetIQ VoIP Assessor
- D. Multi Router Traffic Grapher

ADNV © 14. A company wants to manage its existing IP-enabled PBX system from anywhere within the campus. Which two attributes for the management LAN must you consider when designing the solution? (Choose two)

- A. Security
- B. Autonegotiation
- C. Broadcast traffic
- D. Dynamic Host Configuration Protocol (DHCP)

ADNV 15. A company have a mix of layer 2 and layer 3 switches in its LAN network. Which two QoS methods should you recommend to maintain toll quality VoIP? (Choose two)

- A. 802.1p
- B. Weighted Fair Queuing (WFQ)
- C. Random Early Detection (RED)
- D. Differentiated Services (DiffServ)

ADNV 16. Which tool is best suited for identifying the existing protocols in a customer's network to understand if ... problems may arise between applications and QoS recommendations?

- A. NetAlly
- B. Sniffer Pro
- C. Multi Router Traffic Grapher
- D. Remote Monitoring (RMON)

ADNV 17. The following VoIP network assessment steps have already been completed for a customer's data network

- 1 Estimation of VoIP traffic
- 2 Assessment of LAN/WAN resources
- 3 Capacity is available for VoIP

Which step should you perform next?

- A. Determine network protocol usage
- B. Conduct additional network design analysis
- C. Measure the network's ability to provide QoS
- D. Implement the proposed VoIP network solution

ADNV 18. When assessing a customer's data network for VoIP, why is it important to have a diagram of the network topology?

- A. To determine utilization on the network
- B. To determine the type of CODEC required
- C. To understand the path the voice IP packets will take
- D. To understand the different protocol used on the network

ADNV 19. Before beginning a network assessment of a customer's single site LAN for VoIP, which three questions should you ask the customer? (Choice three)

- A. Which protocols are in use?

- B. Which types of links are in use?
- C. What is the current flow of data and voice traffic?
- D. What is the existing network's ability to support the addition of VoIP traffic with high quality voice?

ADNV 20. In a multisite WAN deployment environment, the challenge is to provide low delay, no jitter and minimal packet loss. This requires a VoIP network assessment that determines the utilization of the routers and links. Which tool is best suited for the task?

- A. Sniffer Pro
- B. HP Openview
- C. Multi Router Traffic Grapher
- D. Remote Monitoring (RMON)

ADNV 21. As part of a VoIP network assessment, you have surveyed the CAT5 wiring to the desktop and have – discover - that the RJ45 data jacks in the wall have only four wires terminated. Which potential issue should you discuss with the customer that may require additional wiring terminations?

- A. Support for 802.1 Q/p
- B. Support for inline power
- C. Support for 100 Base-T Ethernet
- D. Support for shared IP and desktop connections

ADNV 22. During the first phase of a network assessment you discover the customer forces all switch ports to 100 Mbps full duplex. The customer plans to deploy an IP-enable PBX and IP phones. Which network recommendation should you propose to the customer?

- A. Install all devices with default settings
- B. Require all VoIP components to autonegotiate
- C. Force the internet gateways to 100 Mbps full duplex
- D. Force the internet gateways to 100 Mbps full duplex and let the IP phones autonegotiate

VSE 23. Which three H.323 devices have CODECs which may be included in the speech path after a VoIP call-setup?

- A. Terminal
- B. Gateway
- C. Gatekeeper
- D. Multipoint Control Unit (MCU)

VSE 24. A customer requests interoperability with some equipment that uses Megaco signaling and needs to ensure that the VoIP gateways are Megaco compliant. When configuring the VoIP gateways you see a signaling option with four choices, but Megaco is NOT one of them. Which one of the four options should you select to make the products interoperate?

- A. H.225
- B. H.248

- C. H.323
- D. Q.931

VSE 25. You are designing a VoIP network architecture and need to understand the relationships between Media Gateways (MGs) and the Media Gateway Controller (MGCs). Which statement best describes the relationship between MGs and MGCs in the network using MEGACO signaling?

- A. A MG and an MGC can communicate using Megaco or H.323 signaling
- B. A MG and an MGC must communicate using Megaco and cannot use any other protocol
- C. A MG and an MGC cannot communicate using Megaco and must use a protocol like Session Initiation Protocol (SIP)
- D. A MG and an MGC can communicate using Megaco while SIP is being use in the same network

VSE 26. H.323 provides network intelligent and services using gatekeepers. What are the three different network server types used by SIP to provide these services?

- A. A proxy, a gateway, and redirect
- B. A redirect, a proxy, and a registrar
- C. A registrar, a redirect, and an agent
- D. A gateway, an agent, and a director

VSE 27. How does SIP support multimedia sessions for IP telephony?

- A. Uses a media-description language and interworks with web technologies
- B. Is geared toward a particular set of services, including data applications and basic voice/video setup
- C. Uses signaling methods and flows that were patterned upon the ISDN multimedia standards
- D. Provides the transport layer for appropriate applications-session establishment with minimal need for charges to other protocols and applications

* VSE 28. When call setup is established over TCP in an IP-enabled H.323 network, which protocol defines the call-signaling method (direct or gatekeeper routed) that is decided by a gatekeeper during Registration Admission, and Status (RAS) operations?

- × A. G.723
- × B. H.225
- × C. H.263
- D. Q.931

VSE 29. In comparison with H.323 protocol, why is debugging and IP telephony call easier using SIP?

- A. It requires a monitor with abstract syntax notation 1 decoding support
- B. Its signaling is modeled after the HTTP
- * → C. It is executed form any endpoint terminal or device on the network transporting the IP telephony call

- C. Header compression
- D. Payload identification.
- E. Packet retransmission
- F. Enables jitter measurement

PTGK 35. You are drawing a network diagram for a large customer site that has an 802.11b wireless infrastructure using H.323 wireless devices. Through which devices does the call setup traffic flow when making a call to the PSTN?

- A. H.323 wireless device – access point – PSTN
- B. H.323 wireless device – access point – gateway – PSTN
- C. H.323 wireless device – access point – call signaling server – PSTN
- D. H.323 wireless device – access point – call signaling server – media gateway – PSTN

PTGK 36. Given the following bandwidth requirements for G.711 CODEC transported over IP

- 20 ms = 160 bytes of voice payload
- IP packet header is 40 bytes
- 50 IP packets generated for 1 second of voice

What is the expected bandwidth required per call at the IP Layer ignoring Layer 2 media overhead?

$$(160 + 40) \times 8 \times 50 = 80000$$

- A. 40 kbps
- B. 64 kbps
- C. 80 kbps
- D. 96 kbps

PTGK 37. When transporting an IP voice packet over an Asynchronous Transfer Mode (ATM) network, which three specifications required consideration? (Choose three)

- A. ATM adds an additional 10-15% of overhead.
- B. ATM transport requires the frame to be segmented to fit in to multiple cells.
- C. ATM provides a maximum fault tolerance because it is connectionless.
- D. A Continuous Bit Rate (CBR) service can be deployed, providing a dedicated channel/fixed bandwidth for voice.

PTGK 38. Which conversation method is the "benchmark" for voice?

- A. Pulse Code Modulation (PCM) @ 64 kbps
- B. Adaptive Differential Pulse Code Modulation (ADPCM) @ 32 kbps
- C. Low Delay Code – Excited Linear Prediction (LD-CELP) @ 16 kbps
- D. Conjugate – Structure Algebraic Code – Excited Linear Prediction (CS-ACELP) @ 8 kbps

PTGK 39. Given the following customer network information

- IP clients are located a private network of 10.168.254. 0/24 (site A) using Network Address Translation (NAT).
- H.323 gateway is located on the public network at 207.46.197.102 (site B) at a different location.

- D. It requires specialized tools adapted to International Telecommunications Union (ITU) versions

VSE 30. In SIP, signaling allows call information to be carried across network boundaries. What does session management provides?

- A. Handles the transfer and termination of calls
B. The ability to control the attributes of an end-to-end call
C. A fixed duration for end-to-end multimedia communication
D. The predetermined capabilities of the SIP application server

DTGL 31. If you have 256 kbps link, which CODEC should provide the maximum voice channels?

- A. G.711
→ B. G.723.1
C. G.726
D. G.729

DTGL 32. Given the following IP telephony solution components:

- IP terminals
- Media gateways
- Call signaling servers
- H.323 gateways
- Call servers

When one IP user places a call to another IP user on the same system, through which components does the voice media path for the call flow?

- A. IP terminal – IP terminal
X – ~~B. IP terminal – media gateway – IP terminal~~
C. IP terminal – call signaling server – IP terminal
D. IP terminal – call signal. server – call server – call signal. server – IP terminal

DTGL 33. A large text file is send across the LAN/WAN using FTP application. Which statement best describes why FTP would use TCP at Layer 4 of the OSI model to complete the file transfer?

- A. TCP reports on the QoS of the established session
B. TCP allows for multiple files to be transferred at once
C. TCP is a common protocol that routers and switches support easily
→ D. TCP is a connection-oriented protocol where the integrity of the data is more important than the end-to-end transmission time

DTGL 34. The basic attributes of RTP provide for support of applications like voice and video. Which four attributes of RTP support real-time applications? (Choose four)

- A. Timestamping
→ B. Packet sequencing

- The customer network between the IP clients and H.323 gateway appears to be set up properly because you can ping from site A to site B.

A customer is complaining of a one – way speech path in this setup when calling from an IP set to the H.323 gateway. What could be causing the issue?

- A. The IP clients need a gatekeeper to place calls to an H.323 gateway.
- B. The IP clients need a media gateway to place calls to an H.323 gateway.
- C. The H.323 gateway needs a media gateway to transcode the calls to the IP clients.
- D. The IP clients use private address. The media path from the public address cannot reach the IP client with the private address.

PTGK 40. A video conference is in progress between two networked locations using Gigabit Ethernet. The video conferencing application is using User Datagram Protocol (UDP) to transport the video and audio data. Why should UDP be used instead of Transmission Control Protocol (TCP)?

- A. TCP cannot handle streaming data applications.
- B. UDP serves as an efficient transport for handling real – time application traffic.
- C. UDP is better than TCP at seamlessly synching voice and video together.
- D. UDP users less bandwidth than TCP TCP in a WAN, but more bandwidth than TCP.

PTGK 41. Given the following bandwidth requirements for G.729 CODEC transported over IP

- Voice sample size of 10 = 10 bytes
- IP packet header is 40 bytes
- 100 IP packets generated for 1 second of voice

What is the expected bandwidth required per call at the IP Layer ignoring the Layer 2 media overhead?

$$(10 + 40) \times 8 \times 100 = 40 \text{ kbps}$$

- A. 8 kbps
- B. 16 kbps
- C. 24 kbps
- D. 40 kbps

TCI 42. An organization is running voice and data traffic between two sites. They are planning to connect VoIP gateways in IP – enabled Private Branch Exchange (PBX) to existing Ethernet Layer 3 switches at each site. In this situation, which QoS method should achieve the best voice quality using QoS prioritization?

- A. Port Prioritization
- B. IP Address Prioritization
- C. Traffic Separation using VLANs
- D. Differentiated Services (DiffServ)

TCI 43. In order to ensure 99.999% reliability for voice, what must an IP infrastructure provide?

- A. E911 Station Locators.
- B. Data switches and surge protection power strips for IP phones.
- C. Additional power outlets and data connections in the cubicles for IP telephony terminals.
- D. Sufficient cooling in wiring closets for the additional Uninterruptible Power Supply (UPS) and telephony power requirements.

TCI

44. A customer wants to run voice over the WAN to cut down on costs. A combination of a fixed 128 kbps microwave – wireless data connection and Frame Relay circuit join the customer's offshore oil platform to the company network. You determine that a 400 ms delay budget for all VoIP calls on the customer's WAN is acceptable for this type of connection. Which area(s) should you consider for improving the overall delay?

- A. Change from microwave to satellite technology to lower latency.
- B. Nothing can be done because wireless technology has high inherent delay.
- C. Increase the WAN Maximum Transmission Unit (MTU) and bandwidth, and change the company's Frame Relay circuit to Point – to – Point Protocol (PPP).
- D. Decrease the WAN Maximum Transmission Unit (MTU), increase the wireless bandwidth, and change the company's Frame Relay circuit to Point – to – Point Protocol (PPP).

$$S/D = \frac{MTU \times 8}{link}$$

TCI

45. A coworker is engineering an existing data network for VoIP to be added and is trying to figure out how much delay will be inherent in the phone calls.

Given the following information:

- The customer's WAN is a 64 kbps link between sites.
- The Maximum Transmission Unit (MTU) is configured to 1024 bytes.
- The coworker calculates an inherent delay of 60 ms in every call over this data network.

$$S/D = \frac{1024 \times 8}{64 kbps}$$

$$S/D = 128 Kbps$$

What the coworker obviously forgot?

- A. Nothing. The 60 ms delay is correct for this configuration.
- B. The 64 ms speed delay that is inherent in 64 kbps connections.
- C. The 68 ms serialization delay introduced from the WAN configuration.
- D. The 128 ms serialization delay introduced from the WAN configuration.

TCI

46. A customer reports that users are complaining about poor voice quality with VoIP calls. A technician has identified an Ethernet switch in the path for VoIP traffic that appears to be the cause of the poor voice quality. Which two configuration parameters, known to cause problems for VoIP traffic, should be checked on the Ethernet switch? (Choose two)

- A. Duplex setting
- B. Autonegotiation
- C. VLAN configuration
- D. Spanning tree

TCI

47. A campus Enterprise customer has a satellite location a few miles from the campus headquarters. The LAN at the satellite location consists primarily of Layer 2 switching

components. The LAN at the campus headquarters was recently upgraded and uses Layer 3 switching and routing nodes. The customer wishes to deploy VoIP throughout both locations, but wants to segment the voice traffic so that each department within the company can be billed separately. Which QoS method should be used to assure that the voice traffic quality is preserved?

- A. Use Differentiated Services (DiffServ) at both locations
- B. Use packet fragmentation and 802.1Q/p at both locations
- C. Use 802.1Q/p at the satellite location and DiffServ at the headquarters
- D. Use Asynchronous Transfer Mode (ATM) between the satellite and headquarters locations

TCI 48. A customer has a campus network with different vendor switches and has decided to utilize 802.1Q Virtual LANs (VLANs) to implement QoS. Which three methods can be used to create separate VLANs for voice and data traffic? (Choose three)

- A. Protocol
- B. Physical Port
- C. Packetization Rate
- D. Header Compression
- E. Media Access Control (MAC) address

TCI 49. An organization is planning to connect a VoIP gateway from an IP-enabled Private Branch Exchange (PBX) to its existing Ethernet Layer 2 switch. IP phones will be directly attached to the Ethernet Layer 2 switch. In this situation using Layer 2 technology, which two QoS methods should achieve the best quality using QoS prioritization? (Choose two)

- A. Port – based Prioritization
- B. Socket – based Prioritization
- C. Traffic Separation using VLANs
- ~~→~~ D. Differentiated Services (DiffServ)

TCI 50. Which three LAN/WAN network infrastructure requirements for the implementation of VoIP can be impacted by the customer's data traffic? (Choose three)

- A. Echo
- B. Latency
- C. Packet loss
- D. Signaling protocol
- E. Bandwidth availability

TCI 51. VoIP requires a network to provide the following:

- Low delay
- Minimal jitter
- Echo cancellation
- Low packet loss

Which devices should achieve the "best" voice performance if deployed in a network?

- A. Routers with filters at the network core and Layer 2 switches with LANs at the network edge.
- B. Layer 2/Layer 3 switches with 802.1Q at the core network and managed hubs at the network edge.
- C. Layer 2 switches at the core network with Differentiated Services (DiffServ) and routers with 802.1p at the network edge.
- D. Layer 3 switches at the network core with Differentiated Services (DiffServ) and Layer 2 switches with 802.1p at the network edge.

TCI 52. A company recently added VoIP to its data network and is complaining that there is too long of a delay on call over the WAN connection

$$S/D = \frac{1024 \times 8}{56 \text{ kbps}} = 146.29 \rightarrow -10 \quad 136.29$$

Given the following information:

- The WAN is a 56 kbps link
- The Maximum Transmission Unit (MTU) on the WAN is 1024 bytes

$$\text{link} = \frac{1024 \times 8}{136.29}$$

$$\text{link} = 60.11 \text{ kbps}$$

What is the maximum WAN bandwidth required to get the delay introduced from the WAN to the dropped down to under 10 ms without changing the MTU?

- A. increase the WAN bandwidth to 64 kbps
- B. increase the WAN bandwidth to 128 kbps
- C. increase the WAN bandwidth to 256 kbps
- D. it cannot be done as second delays are normal for VoIP technology

Voice over Internet Protocol Technologies Practices

Practice 2: Packet Telephony Overview

Answer the following questions:

1. A customer requires the least amount of bandwidth usage and is not concerned with voice quality. Which CODEC would you recommend?
 - a. G.711
 - b. G.729A
 - c. G.723.1 →
 - d. G.726
2. A customer requires the best balance of quality audio and bandwidth savings. Which CODEC would you recommend?
 - e. G.711
 - f. G.729A
 - g. G.723.1
 - a. G.727
3. At what rate is an analog sampled using a G.711 CODEC?
 - a. 2 kHz
 - b. 3 kHz
 - c. 4 kHz
 - d. 8 kHz
4. What are the two main parts of a voice packet?
 - a. Header and Payload
 - b. Leader and Header
 - c. Footer and Load
 - d. Leader and Message
5. On which layer of the OSI model does IP reside?
 - a. Layer 1
 - b. Layer 2
 - c. Layer 3
 - d. Layer 5
6. Which Layer 2 protocol transmits data at a fixed size unit called a cell?
 - a. PPP
 - b. ATM
 - c. PPTP
 - d. CBR
7. Given the following parameters:
 - G.711 CODEC
 - 10 ms Voice sample
 - IP packet header is 40 bytesIgnoring Layer 2 information, what is the expected bandwidth required per call?
 - a. 256 kbps

- b. 128 kbps
- c. 96 kbps
- d. 80 kbps

8. Given the following parameters:

- G.729 CODEC
- 20 ms Voice sample
- IP packet header is 40 bytes

Ignoring Layer 2 information, what is the expected bandwidth required per call?

- a. 96 kbps
- b. 72 kbps
- c. 48 kbps
- d. 24 kbps

9. Which Protocol is highly reliable, connection oriented and best suited for situations when the integrity of the data is more important than the end-to-end transmission time.

- a. UDP
- b. TCP
- c. RTP
- d. PPP

10. Which protocol is designed to monitor the Quality of Service and to convey information about participants in a real-time session?

- a. RTCP
- b. IP
- c. TCP
- d. SIP

Practice 5: VoIP Standardization and Signalling Protocols

Answer the following questions:

1. Within the H.323 protocols, the standard that defines control parameters is _____.

- a. H.261
- b. G.729
- c. H.225
- d. H.245

2. A key advantage of utilizing H.323 is that H.323 provides a guaranteed QoS.

- a. True
- b. False

3. When programming a codec in an H.323 device for voice transmission on an IP WAN network, _____ is the assigned default.

- a. G.711
- b. G.723.1
- c. G.728
- X → d. G.729

4. The first major operation in establishing an H.323 call setup is _____.

- a. RAQ
- b. ACF
- c. ARJ
- d. ACK

- e. RAQ
- f. ACF
- g. ARJ
- h. ACK

5. A key function of the SIP Redirect Server is the ability to _____.

- a. Forward SIP requests to other servers
- b. Accept REGISTER requests
- c. Offer location services
- d. Map an address in the request to a new address, returning the message to the client

6. The first major operation in SIP call setup is _____.

- a. DISCOVER
- b. REQUEST
- c. ACK
- ⇒ d. INVITE

7. A SIP Response Type indicating a successful connection is _____.

- a. 1xx
- b. 2xx
- c. 3xx
- d. 4xx

8. The role of MGCP is to _____.

- a. Decompose a Telephony Gateway into controlling signalling and controlled media components
- b. Provide high availability in the event of equipment failures
- c. Communicate to the MG and SG via TCP/IP
- d. Support Proxy Servers

9. A function of MEGACO MGC is to provide control and intelligence to the MG.

- a. True
- b. False

